REMARKS

By this amendment, claims 3, 7, 10 and 14 have been cancelled and claims 1, 4, 12 and 18 have been amended. Currently, claims 1-2, 4-6, 8-9, 11-13 and 15-19 are pending in the application.

The Examiner stated that claim 12 was objected to and the Examiner suggested deleting "step of" from "said step of re-sampling" on line 15 of claim 12 to avoid improper antecedent basis issues. By this amendment, the term "step of" has been deleted from claim 12 as the Examiner suggested. It is respectfully submitted that this objection should be withdrawn in view of this amendment.

Claims 1-19 were rejected under 35 USC 103(a) as being obvious over Otomo et al. (U.S. Patent No. 6,580,671) in view of Zhang (U.S. Patent No. 6,295,362). The Examiner believed that Zhang discloses a FIR filter is used to suppress aliasing distortion and teaches that by using the FIR filter, aliasing distortion due to re-sampling is suppressed. The Examiner also believed that it would have been obvious to incorporate into the filter of Otomo et al. the FIR filter of Zhang, the motivation being to suppress aliasing distortion created during the re-

sampling of the first audio data of Otomo et al. The Examiner also believed that regarding claims 3 and 14, the signal processing delay time in said filter will inherently correspond to a predetermined processing unit of inputted audio data in some manner.

This rejection is respectfully traversed in view of the amendments to the claims 1 and 12 and the following remarks.

The present invention relates to a signal processing device and method which perform signal processing for reproducing multi-channel audio data been recorded on an optical disk or the like. In the case of a DVD, the information capacity is greater than 4 GB, and an application such as a movie is recorded on the DVD. The information read out rate of a DVD player is about 10 Mbps. Although it is desired to employ a high sampling rate for the audio signal in order to achieve high sound quality, if the sound is in multi-channel format, the reading out rate can exceed the above value.

The present invention disclosed a data stream D3 which has been encoded by an encoding circuit, like the encoding circuit 5 shown in Fig. 1, is input via the terminal 7 to the decoding circuit 16. The decoding circuit 16 extracts the header data from the headers of this data stream D3, and separates the data

stream D3 into a first audio data D1 which corresponds to the sampling frequency fs1 of 48 kHz or 44.1 kHz, and a second audio data D2 which corresponds to the sampling frequency of fs2(=2×fs1) of 96 kHz or 88.2 kHz. The first audio data D1 is input to the buffer 9, while the second audio data D2 is input to the buffer 10.

In this manner the following procedures are executed for each block in the data stream D3: (1) the headers are removed; (2) it is separated into the two data D1 and D2; and (3) these audio data D1 and D2 are output to the buffers. In other words, these procedures are performed for each group of 40 samples of the first audio data D1 and for each group of 80 samples of the second audio data D2. The first and second audio data D1 and D2 which have been output for every one block from this decoding circuit 16 are respectively input to the buffers 9 and 10.

Next the output data from the buffer 9 is input to the up-sampling circuit 17. The up-sampling circuit 17 performs up-sampling upon this input data at approximately twice the sampling frequency, in consideration of the number of data elements (in this case 80) in one block of the second audio data D2. If for example the FIR filter 17b is used in the up-sampling circuit 17, then its delay amount will be equal to (N+1)/2 in terms of the

tap number N of the FIR filter 17b. Accordingly, if the number of data elements in the second audio data D2 is supposed to be 80, then the characteristics of the FIR filter 17b are set so that its tap number N should be equal to 159, in order for this delay amount to become adequate for 80 samples.

On the other hand, the output data elements from the buffer 10 are input to the delay buffer 18. This delay buffer 18 generates the delay period of one block by storing the number of data elements in one block (= 80 data elements). The delay buffer 18 has a structure as shown in Figs. 7A and 7B. Fig. 7A shows the input-output state at a certain time point, while Fig. 7B shows the input-output state at the next clock signal. When one data N1 is input, the delay buffer 18 outputs the data element 01 among previously stored data elements 01 to 80. Actually, the input data N1 is input at a clock rate which is synchronized with the sampling frequency. In this manner, the delay amount of the signal is easily controlled by making the tap number of the FIR filter in the up-sampling circuit 17 equal to the number of data elements in one block of audio data which is input.

Independent claims 1 and 12 have both been amended to recite "wherein signal processing delay time in said filter corresponds

to a predetermined processing unit of inputted audio data". The features in claims 1 and 12 are not shown or suggested by Otomo et al. or Zhang.

Otomo et al. relate to a digital audio recording medium and a reproducing apparatus. Otomo et al. disclose that high-density recording optical disks on which the main picture signal, plural types of sub-picture signals accompanying the main picture signal, and audio signals of plural channels can be recorded, have been developed.

Otomo et al. also disclose that the signal read optically from the disk 18 passes through a demodulating section and is input to a packet processing section 21. The demodulating section performs an error correction process and a modulating process. The packet processing section 21 identifies a channel group, referring to the identifier of the packet header. The identification discriminates between the packet in the first channel group and the packet in the second channel group.

Otomo et al. also disclose that the signal in the first channel group is input to a frame processing section 22, which cancels the frame and outputs an R channel signal and an L channel signal. The signal in the second channel group is input

to a frame processing section 23, which cancels the frame and outputs C, SR, SL, and SW channel signals.

Otomo et al. also disclose that the R and L channel signals are input to a phase matching section 24. Further, the C, SR, SL, and SW channel signals are input to a frequency converting section 25, which up-converts the sampling frequency fs of 48 kHz into 96 kHz.

However, Otomo et al. do not disclose that a filter for performing re-sampling upon the first audio data at the same sampling frequency fs2 as that of the second audio data, and suppressing aliasing distortion due to the re-sampling, and for outputting the first audio data from said filter as claimed in claims 1 and 12.

Otomo et al. also do not disclose that a signal processing delay time in the filter corresponds to a predetermined processing unit of inputted audio data as claimed in claim 1.

Otomo et al. also do not disclose that a processing period in the step of filtering corresponds to a predetermined processing unit of inputted audio data as claimed in claim 12.

Applicant respectfully submits that Otomo et al. do not disclose the way to select the processing within the decoder

corresponds to the delay time of the re-sampling filter as claimed in claims 1 and 12.

For these reasons, it is believed that Otomo et al. do not show or suggest the presently claimed features of the present invention. Applicants also submit that Zhang does not make up for the deficiencies in Otomo et al.

Zhang relates to the generation of composite stereo signals for broadcasting in the FM frequency band. More particularly, Zhang relates to a novel circuit for direct digital synthesis of composite stereo signals for broadcast in the FM frequency band.

Zhang discloses an anti-aliasing filter, which is preferably a 40-tap FIR low pass filter 405 is used to remove the aliasing components.

However, Zhang does not disclose that signal processing delay time in the filter corresponds to a predetermined processing unit of inputted audio data as claimed in claim 1.

Zhang also does not disclose that a processing period in the step of filtering corresponds to a predetermined processing unit of inputted audio data as claimed in claim 12.

Applicants respectfully submit that FIR filter in Zhang would not be combined with the frequency converting section in Otomo et al. because the frequencies of Otomo et al. and Zhang

are very different. Otomo et al. relate to a digital audio recording medium and a reproducing apparatus (for example, the high-density recording optical disk such as a DVD). Zhang relates to the generation of composition stereo signals for broadcasting in the FM frequency band. Both of the frequencies used in Otomo et al. and Zhang are very different. Also, FIR filter of the present invention is 159 taps and the anti-aliasing filter of Zhang is 40 taps. Therefore, a FIR filter used for a high-density recording optical disk and a FIR filter used for FM signals are very different and one of ordinary skill in the art would not seek to combine such different technologies.

Moreover, applicants respectfully submit that if the FIR filter in Zhang were required in Otomo et al., the FIR filter would require great computational resources, a large silicon area and high power; therefore, these features teach away from such a combination.

It is respectfully submitted that Otomo et al. and Zhang, individually or in combination, do not teach, disclose or suggest the presently claimed invention and it would not have been obvious to one of ordinary skill in the art to combine these references to render the present claims obvious.

In view of foregoing claim amendments and remarks, it is respectfully submitted that the application is now in condition for allowance and an action to this effect is respectfully requested.

If there are any questions or concerns regarding the amendments or these remarks, the Examiner is requested to telephone the undersigned at the telephone number listed below.

Respectfully submitted,

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